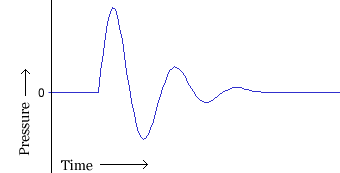
**Tutorial - I.Basics  
Part 1 -  Digital Audio  - Part 1**

What is sound?

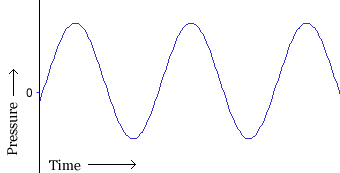
Sounds are pressure waves of air. If there wasn't any air, we wouldn't be able to hear sounds. There's no sound in space.

We hear sounds because our ears are sensitive to these pressure waves. Perhaps the easiest type of sound wave to understand is a short, sudden event like a clap. When you clap your hands, the air that was between your hands is pushed aside. This increases the air pressure in the space near your hands, because more air molecules are temporarily compressed into less space. The high pressure pushes the air molecules outwards in all directions at the speed of sound, which is about 340 meters per second. When the pressure wave reaches your ear, it pushes on your eardrum slightly, causing you to hear the clap.



A hand clap is a short event that causes a single pressure wave that quickly dies out. The image above shows the waveform for a typical hand clap. In the waveform, the horizontal axis represents time, and the vertical axis is for pressure. The initial high pressure is followed by low pressure, but the oscillation quickly dies out.

The other common type of sound wave is a periodic wave. When you ring a bell, after the initial strike (which is a little like a hand clap), the sound comes from the vibration of the bell. While the bell is still ringing, it vibrates at a particular frequency, depending on the size and shape of the bell, and this causes the nearby air to vibrate with the same frequency. This causes pressure waves of air to travel outwards from the bell, again at the speed of sound. Pressure waves from continuous vibration look more like this:



How is sound recorded?

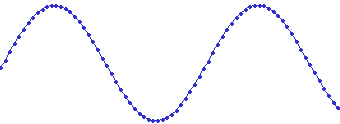
A microphone consists of a small membrane that is free to vibrate, along with a mechanism that translates movements of the membrane into electrical signals. (The exact electrical mechanism varies depending on the type of microphone.) So acoustical waves are translated into electrical waves by the microphone. Typically, higher pressure corresponds to higher voltage, and vice versa.

A tape recorder translates the waveform yet again - this time from an electrical signal on a wire, to a magnetic signal on a tape. When you play a tape, the process gets performed in reverse, with the magnetic signal transforming into an electrical signal, and the electrical signal causing a speaker to vibrate, usually using an electromagnet.

How is sound recorded digitally ?

Recording onto a tape is an example of analog recording. Audacity deals with digital recordings - recordings that have been sampled so that they can be used by a digital computer, like the one you're using now. Digital recording has a lot of benefits over analog recording. Digital files can be copied as many times as you want, with no loss in quality, and they can be burned to an audio CD or shared via the Internet. Digital audio files can also be edited much more easily than analog tapes.

The main device used in digital recording is a Analog-to-Digital Converter (ADC). The ADC captures a snapshot of the electric voltage on an audio line and represents it as a digital number that can be sent to a computer. By capturing the voltage thousands of times per second, you can get a very good approximation to the original audio signal:



Each dot in the figure above represents one audio *sample*. There are two factors that determine the quality of a digital recording:

* **Sample rate**: The rate at which the samples are captured or played back, measured in Hertz (Hz), or samples per second. An audio CD has a sample rate of 44,100 Hz, often written as 44 KHz for short. This is also the default sample rate that Audacity uses, because audio CDs are so prevalent.
* **Sample format** or **sample size**: Essentially this is the number of digits in the digital representation of each sample. Think of the sample rate as the horizontal precision of the digital waveform, and the sample format as the vertical precision. An audio CD has a precision of 16 bits, which corresponds to about 5 decimal digits.

Higher sampling rates allow a digital recording to accurately record higher frequencies of sound. The sampling rate should be at least twice the highest frequency you want to represent. Humans can't hear frequencies above about 20,000 Hz, so 44,100 Hz was chosen as the rate for audio CDs to just include all human frequencies. Sample rates of 96 and 192 KHz are starting to become more common, particularly in DVD-Audio, but many people honestly can't hear the difference.

Higher sample sizes allow for more dynamic range - louder louds and softer softs. If you are familiar with the decibel (dB) scale, the dynamic range on an audio CD is theoretically about 90 dB, but realistically signals that are -24 dB or more in volume are greatly reduced in quality. Audacity supports two additional sample sizes: 24-bit, which is commonly used in digital recording, and 32-bit *float*, which has almost infinite dynamic range, and only takes up twice as much storage as 16-bit samples.

Playback of digital audio uses a Digital-to-Analog Converter (DAC). This takes the sample and sets a certain voltage on the analog outputs to recreate the signal, that the Analog-to-Digital Converter originally took to create the sample. The DAC does this as faithfully as possible and the first CD players did only that, which didn't sound good at all. Nowadays DACs use Oversampling to smooth out the audio signal. The quality of the filters in the DAC also contribute to the quality of the recreated analog audio signal. The filter is part of a multitude of stages that make up a DAC.

How does audio get digitized on your computer?

Your computer has a soundcard - it could be a separate card, like a SoundBlaster, or it could be built-in to your computer. Either way, your soundcard comes with an Analog-to-Digital Converter (ADC) for recording, and a Digital-to-Analog Converter (DAC) for playing audio. Your operating system (Windows, Mac OS X, Linux, etc.) talks to the sound card to actually handle the recording and playback, and Audacity talks to your operating system so that you can capture sounds to a file, edit them, and mix multiple tracks while playing.

Standard file formats for PCM audio

There are two main types of audio files on a computer:

* PCM stands for Pulse Code Modulation. This is just a fancy name for the technique described above, where each number in the digital audio file represents exactly one sample in the waveform. Common examples of PCM files are **WAV** files, **AIFF** files, and **Sound Designer II** files. Audacity supports WAV, AIFF, and many other PCM files.
* The other type is compressed files. Earlier formats used logarithmic encodings to squeeze more dynamic range out of fewer bits for each sample, like the u-law or a-law encoding in the **Sun AU** format. Modern compressed audio files use sophisticated psychoacoustics algorithms to represent the essential frequencies of the audio signal in far less space. Examples include **MP3** (MPEG I, layer 3), **Ogg Vorbis**, and **WMA** (Windows Media Audio). Audacity supports MP3 and Ogg Vorbis, but not the proprietary WMA format or the MPEG4 format (AAC) used by Apple's iTunes.

For details on the audio formats Audacity can import from and export to, please check out the [Fileformats page](http://audacity.sourceforge.net/manual-1.2/fileformats.html) of this documentation. Please remember that MP3 does not store uncompressed PCM audio data. When you create an MP3 file, you are deliberately losing some quality in order to use less disk space.

## Part 2 -  Rules of Audacity  - Part 2

If you'd like to get straight playing an imported file or recording something, you can skip this section and come back later.

Whenever you work with Audacity, there are some rules you should remember:

1. One clip per track

A clip is simply a piece of audio material. Imported, recorded, split or duplicated from another track, one track can only carry one piece of audio at a time. You can extend it by pasting material or inserting silence in to it, or cut a piece away, but it will always be one continuous piece of audio.

2. Audacity always records to a new track

This new track is opened at the bottom. You'll have to zoom out and then resize the track view of the bottom most track to see what is recorded. You can actually use the window sliders at the bottom and right to do this after starting to record, but this way no performance will be lost to the windowing system.  
I suggest hitting CTRL+F to get an overview of the entire project as well. This only affects the horizontal zoom by the way(left-right zoom). There is no way to zoom out vertically without using the mouse yet.

3. Edit/Duplicate will not create a new audio file

This may not seem a big deal, but it is if you're editing a large live recording.

What Audacity does is reference the original audio material until you actually perform some kind of edit on it, such as cutting a piece away, or using any effect on it. One thing to remember is the **UNDO** function. You can undo/redo stuff as many times as you like, and yes, even after you have saved your project.

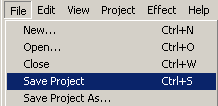
You may ask what happens if you do, for example, cut away a piece or mark off a 30 minute piece and split it to a new track. It only writes changed data to disk. Since Audacity works with chunk of audio data of around one megabyte in size, this happens quite fast. Rest assured that the only big waiting period might be the importing of large audio files.

**Part 3 -  Setup, Audio Import and Playback  - Part 3**

1. Create a new project

This is very important!

Audacity writes all the changed and recorded audio to a directory called ***Projectname*\_data**, which is located right where you saved the project file itself.

Thus, select and choose a location and filename for your project.

Please note that when you startup Audacity fresh, only the **" Save As..."** menu option is available.

To save your project later on, you can also use the keyboard shortcut : CTRL+S

2. Check the Preferences

Again, this is very important!

|  |  |
| --- | --- |
| Press CTRL+P or go to ... Preferences ... | ...then check if the right output is selected :  Audio I/O tab of  the Preferences |

|  |  |
| --- | --- |
| ...set the sample rate of your choice... (44.1 kHz is the default) Quality tab of the Preferences | ...and here's a crucial screen : File Formats tab of the Preferences |

The *File Formats* settings need discussing at this point.

***When importing uncompressed audio***, there are two ways to do it. "*Make a copy of the original before editing*" means, that Audacity actually copies the entire audio file that you imported in to its project data directory and in the process sets up the little overview graphics, whose descriptions get stored in the project data directory too.

The second way is to use the original imported audio. You may think we're actually editing this file, but no we aren't. In fact, Audacity will now read the imported file once and simply create the graphics overviews for them in the data directory, and subsequently write to disk all the audio data that you change. The original file is only used for playback. All audio that remains unchanged will be played from the original file.

The advantage of choosing to **make a copy of the original** is that you avoid trouble, should **anything** in the original file change.

For example, should you accidentally delete the original file, you're lost.

You have to make up your mind before you start a project. Choose to make a copy of all imported files, and you'll use more space on your hard disk(s), but it will be easier to back up the project too, because all files that have anything to do with your project will be in the project data directory.

The *Uncompressed Export Format* can be set to WAV or AIFF for now. Please check the [fileformats page](http://audacity.sourceforge.net/manual-1.2/fileformats.html) for further information on export formats.

We'll ignore the *Spectrogram* settings for now. The *Directories* setting can be ignored as well for now, because all it sets is the directory to use for recordings, undo data and other stuff, if you haven't yet saved your project. Since we already saved our project, this setting is of no importance to us, though you may want to set it properly later on. Initially this is set to a folder called "audacity\_temp\_1.2" in the system temporary directory.

3. Import an audio file

There are three ways to do this:

1. Simply drag and drop the audio file in to the Audacity window. (If you're using Mac OS 9 or X, drag the audio file to the Audacity icon instead...)

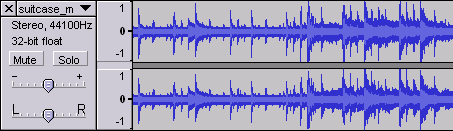
2. Select *Import Audio ...* in the Project menu.

3. Use the keyboard shortcut : CTRL+I

Audacity can import WAV, AIFF, AU, IRCAM, MP3 and OGG files. Please refer to the [fileformats page](http://audacity.sourceforge.net/manual-1.2/fileformats.html) for further reference on these audio formats.

4. Playback

The imported file should now be displayed in an audio track. The track will look a little like this, depending on what you imported :

  
Trackpanel and Waveform Overview of the imported Track

If you're not sure where to find audio material, simply rip some off a CD, or in Windows, check the Media folder in the directory of your Windows installation.

Now click on the green Play button [Play Button](http://audacity.sourceforge.net/manual-1.2/toolbar.html#play)at the top and you should hear the file you have just imported.

**Part 4 -  Recording with Audacity   - Part 4**

1. Create a new project

Save an empty project. Or simply use the one from the [previous part](http://audacity.sourceforge.net/manual-1.2/tutorial_basics_3.html). Remember, that if you don't save your project before you start recording or importing, that all recordings, edit and other files will be written to the directory set on the [*Directories*](http://audacity.sourceforge.net/manual-1.2/prefs.html#directories) tab of the [preferences](http://audacity.sourceforge.net/manual-1.2/prefs.html).

2. Check the preferences

Make sure your playback and recording device are set. If you're going to record a stereo signal, set the number of channels to record to 2 (Stereo) on the [*Audio I/O*](http://audacity.sourceforge.net/manual-1.2/prefs.html#audioio) preferences.

When picking a device to record from, make sure you've set up all the connections properly, such as plugging a microphone in to the **Mic Input**, and any other device in to the **Line In** of your sound card. Then check that the gain level knob(the amount by how much the input should be amplified) of the mixer of your soundcard is set right.

Since most soundcards can mix the inputs back in to the outputs, the easiest way to test your microphone is to speak in to it while playing with your sound card mixer. The sound card mixer is a piece of software either provided by the sound card maker, or by the operating system you're using. The Windows mixer is pretty straight forward, though some soundcards bring their own along. The Mac's mixer is controlled via the Sound Control Panel, and the Linux users have a variety of mixer applications at their disposal. Just make sure they work before yelling at your screen that nothing works.

3. Hit Record

|  |  |  |
| --- | --- | --- |
| Click on the red **Record** button | [Record](http://audacity.sourceforge.net/manual-1.2/toolbar.html#record) | to begin recording. |
| Click on the blue **Pause** button | [Pause](http://audacity.sourceforge.net/manual-1.2/toolbar.html#pause) | to pause the recording. Press it again to continue. |
| Click on the yellow **Stop** button | [Stop](http://audacity.sourceforge.net/manual-1.2/toolbar.html#stop) | to cease recording. The cursor will return to its previous position, before the recording was started. |

That's it. You can now play around with your recording and explore the editing capabilities of Audacity. Remember that you can use the Undo function almost without limits whilst the project is open.

For more information see the on-line tutorial using the link below:

<http://audacity.sourceforge.net/manual-1.2/tutorials.html>